

**AMENDMENT(S) TO THE CLAIMS**

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4       1.       (original)       A method for transmitting a mixed media data stream in  
5 packets, including audio and video objects, between a sender and a receiver through a  
6 connection over a network, the method comprising:  
7       monitoring, at the receiver, transmission characteristics of the connection between  
8 the server and the receiver;  
9       estimating available bandwidth at the sender based upon the transmission  
10 characteristics of the connection monitored at the receiver;  
11       allocating a global buffer for the mixed media data stream to be transmitted from  
12 the sender to the receiver as a function of the estimated available bandwidth at the sender;  
13       pre-encoding a portion of each Video Object Plane (VOP) in the global buffer  
14 with respect to a quantization parameter (QP) of the VOP;  
15       encoding the VOP in the global buffer based on the QP;  
16       updating a rate distortion model based upon the QP and packet loss rate;  
17       performing a frame skipping function after the VOP encoding; and  
18       transmitting from the sender to the receiver the encoded video object plane in the  
19 global buffer at a regulated sender transmission rate from the sender as a function of the  
20 estimated available bandwidth at the sender.  
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2. (original) The method as defined in Claim 1, further comprising:  
receiving the encoded video object plane at the receiver from the connection;  
demultiplexing the encoded video object plane into coded video and audio  
streams;  
inputting the coded video and audio streams, respectively, into video and audio  
decoders;  
inputting the decoded video and audio streams to a media mixer; and  
inputting the mixed video and audio streams output from the media mixer to an  
output device.

3. (original) The method as defined in Claim 1, wherein pre-encoding a  
portion of each VOP with respect to the QP of the VOP further comprises adjusting the  
QP of the VOP.

4. (original) The method as defined in Claim 3, wherein the QP of the  
VOP is adjusted with respect to a texture parameter,  $r$ , as the number of bits which will be  
used to encode the VOP wherein:

$$r = \frac{p_1 \times MAD}{QP} + \frac{p_2 \times MAD}{QP^2};$$

$p_1$  and  $p_2$  are control parameters; and

MAD is a mean absolute distortion in which a total target bit rate for all objects in  
the global buffer are allocated proportionally to motion, size, and square of MAD.

5. (original) The method as defined in Claim 4, wherein the adjusting of the QP of the VOP is performed by changing the QP to a values in a range from 1 to 31 depending upon the estimated available bandwidth at the receiver.

6. (original) The method as defined in Claim 1, wherein updating a rate distortion model based upon the QP and packet loss rate comprises:

predicting the number of bits,  $r_i$ , to encode the  $i$ th VOP, and is given by:

$$r_i = \frac{(p_1)_i \times MAD_i}{QP_i} + \frac{(p_2)_i \times MAD_i}{QP_i^2};$$

the distortion,  $d_i$ , is estimated by

$$d_i = (q_1)_i \times QP_i + (q_2)_i \times QP_i^2 + (q_3)_i \times r_i \times (P_L)_i, \text{ wherein:}$$

$q_1$ ,  $q_2$  and  $q_3$  are control parameters; and

the packet loss rate  $(P_L)_i$  is an estimate of that the probability that the  $i$ th transmission of data from the sender will be lost; and

minimizing the overall distortion,  $D$ , for each encoded VOP by  $D = \sum_i d_i$ , subject

to  $R = \sum_i r_i \leq R_T$ , where  $R_T$  is the total bit budget for the current time instant.

7. (original) The method as defined in Claim 1, wherein:

the sender sends data to the receiver in through a connection over a packet switched network in a sender packet having a sender header that includes:

- a packet sequence number;
- a timestamp indicating the time when the sender packet was sent (ST1); and
- the size of the sender packet (PacketSize);

the receiver sends data to the sender through the connection over the packet switched network in a receiver packet having a receiver header that includes:

- the time interval that the sender packet spent in the receiver side ( $\Delta RT$ );
- the timestamp of the sender packet sent from the sender (ST1);
- an estimate, calculated by the receiver, of a packet-loss rate; and
- the rate at which data is received at the receiver;

monitoring transmission characteristics of the connection between server and receiver comprises:

- estimating a round trip time of the sender packet from the sender to the receiver (RTT) based on ST1 and  $\Delta RT$ ;
- estimating a time out interval (TO) before which the sender should retransmit to the receiver a sender packet of data that has not been received by the receiver;
- estimating a probability that a packet of data will be lost ( $P_L$ );
- estimating the present available network bandwidth at which the receiver can receive data from the sender (rvcrate) as a function of the PacketSize, the RTT, the  $P_L$ , and the TO;

deriving the present sending rate of data from the sender to the receiver  
 ( $\overline{currate}$ );

setting an updated sending rate of data from the sender to the receiver  
 ( $\overline{currate}$ ), wherein:

if  $\overline{rcvrate}$  is greater than  $\overline{currate}$ , then deriving  $\overline{currate}$  as a  
 function  $\overline{currate}$ , PacketSize, and RTT; and  
 if  $\overline{rcvrate}$  is not greater than  $\overline{currate}$ , then setting  $\overline{currate}$  to  
 be less than  $\overline{rcvrate}$ .

8. (original) The method as defined in Claim 7, wherein:

$$RTT = \alpha \times \overline{RTT} + (1 - \alpha) \times (now - ST1 - \Delta RT); \text{ and}$$

$now$  is the timestamp indicating the time at which the receiver packet was  
 received in the sender; and  $\alpha$  is a weighting parameter.

9. (original) The method as defined in Claim 7, wherein:

$$TO = RTT + (k \times RTTVAR);$$

$k$  is a constant;

$$RTTVAR = \alpha_2 \times \overline{RTTVAR} + (1 - \alpha_2) \times |RTT - (now - ST1 - \Delta RT)|;$$

$\overline{RTTVAR}$  is the current variation in the round trip time of the sender packet from  
 the sender to the receiver (RTT);

$\alpha_2$  is a weighting parameter; and

RTTVAR is a smoothed estimate of  $\overline{RTTVAR}$ .

10. (original) The method as defined in Claim 7, wherein  $P_L$  is derived by a Gilbert Model.

11. (original) The method as defined in Claim 10, wherein:

$$P_L = \frac{\hat{q}}{\hat{p} + \hat{q}};$$

$$\{X_i\}_{i=1}^n;$$

$X_i$  takes 1 if the  $i$ th sender packet has arrived successfully at the receiver;

$X_i$  takes 0 if the  $i$ th sender packet is lost;

$$p = P[X_i = 1 | X_{i-1} = 0];$$

$$q = P[X_i = 0 | X_{i-1} = 1];$$

$\hat{p}$  is an estimate of  $p$ ;

$\hat{q}$  is an estimate of  $q$ ; and

$$\hat{p} = n_{01}/n_0 \text{ and } \hat{q} = n_{10}/n_1, \text{ wherein:}$$

$n_{01}$  is the number of times in an observed time series when one follows zero;

$n_{10}$  is the number of times when zero follows one;

$n_0$  is the number of zeros; and

$n_1$  is the number of ones.

12. (original) The method as defined in Claim 11, wherein:

the  $P_L$  is further smoothed by a filter that weights the  $n$  most recent measured packet loss rates by:

$$P_{L,i} = \sum_{j=0}^{n-1} (w_j \times \overline{P_{L,i-j}})$$

$\overline{P_{L,i-j}}$  is the measured packet loss rate in the  $(i-j)$ th time interval;

two set of weighting parameters are defined as follows:

	W0	W1	W2	W3	W4	W5	W6	W7
WS1	1.0	1.0	1.0	1.0	0.8	0.6	0.4	0.2

	W0	W1	W2	W3	W4	W5	W6	W7
WS2	1.2	1.2	1.0	1.0	0.8	0.5	0.3	0.1

; and WS2 is used for  $w_j$  when the actual packet loss rate is less than half of the measured packet loss rate, otherwise WS1 is used for  $w_j$ .

13. (original) A computer-readable media comprising computer-executable instructions for performing the method as recited in Claim 1.

14. (original) A method for transmitting a mixed media data stream in packets, including audio and multiple video objects (MVOs), between a sender and a receiver through a connection over a network, the method comprising:

- monitoring transmission characteristics of one or more encoded video object planes through the connection between the sender and the receiver;
- estimating, from the transmission characteristics, an available bandwidth ( $R_T$ ) at the sender;
- allocating, as a function of the  $R_T$ , a portion of the mixed media data stream to a global buffer;
- encoding a video object plane from the global buffer based upon a rate distortion function that accounts for packet loss rate between sender and receiver;
- updating the rate distortion function based upon results of the encoded video object plane and upon a memory containing results of one or more previously encoded video object planes;
- after the encoding the MVOs in the video object plane, performing a frame skipping function; and
- transmitting, at the estimated available bandwidth, the encoded video object plane from the sender to the receiver.



15. (original) The method as defined in Claim 14, wherein allocating a portion of the mixed media data stream to a global buffer comprises:

$W_{cur} = \max(((W_{prev} + B_{prev}) \times R_T / R_{old} - R_T / F), 0)$ , as the global buffer size  $R_{old}/2$ , is changed to  $R_T/2$ , wherein:

$B_{prev}$  is the number of bits spent in the previous time instant  $B_{prev}$ ,

$R_{old}/2$  is the previous size of the global buffer;

$W_{prev}$  is the previous occupancy of the global buffer; and

$F$  is the video frame rate.

16. (original) The method as defined in Claim 14, wherein allocating a portion of the mixed media data stream to a global buffer comprises the allocation of an output target rate from the global buffer among each of video and audio data streams so as to yield the target bits for an individual object in the data stream.

1 17. (original) The method as defined in Claim 14, further comprising:  
2 receiving the encoded video object plane at the receiver from the connection;  
3 demultiplexing the encoded video object plane into coded video and audio  
4 streams;  
5 inputting the coded video and audio streams, respectively, into video and audio  
6 decoders; and  
7 inputting the decoded video and audio streams to a media mixer; and  
8 inputting the mixed video and audio streams output from the media mixer to an  
9 output device.  
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11 18. (original) A computer-readable media comprising computer-  
12 executable instructions for performing the method as recited in Claim 14.  
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15 19. (canceled).  
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19 21. (canceled).  
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33. (original) One or more computer-readable media, comprising stored thereon:

- a first set of elements to describe a server in communication through a connection of a packet switched network to a client;
- a second set of elements to describe the transmission of a mixed media data stream, including audio and multiple video objects (MVOs), from the server to the client through a connection over a packet switched network; and
- a code segment that, when executed:
  - monitors transmission characteristics of the connection between server and receiver;
  - estimates available bandwidth at the sender based upon the monitored transmission characteristics of the connection;
  - allocates a global buffer for the mixed media data stream to be transmitted stream from the sender to the receiver as a function of the estimated available bandwidth at the sender;
  - pre-encodes a portion of each Video Object Plane (VOP) in the global buffer with respect to a quantization parameter (QP) of the VOP;
  - encodes the VOP in the global buffer based on the QP;
  - updates a rate distortion model based upon the QP and packet loss rate;
  - performs a frame skipping function after the VOP encoding; and
  - transmits from the sender to the receiver the encoded video object plane in the global buffer at a regulated sender transmission rate from the sender as a function of the estimated available bandwidth at the sender.

1           34.   (original)   One or more computer-readable media, comprising stored  
2   thereon:  
3           a first set of elements to describe a server in communication through a connection  
4   of a packet switched network to a client;  
5           a second set of elements to describe the transmission of a mixed media data  
6   stream, including audio and multiple video objects (MVOs), from the server to the client  
7   through a connection over a packet switched network; and  
8           a code segment that, when executed:  
9           monitors transmission characteristics of one or more encoded video object  
10   planes through the connection between the sender and the receiver;  
11           estimates, from the transmission characteristics, an available bandwidth  
12   ( $R_T$ ) at the sender;  
13           allocates, as a function of the  $R_T$ , a portion of the mixed media data stream  
14   to a global buffer;  
15           encodes a video object plane from the global buffer based upon a rate  
16   distortion function that accounts for packet loss rate between sender and receiver;  
17           updates the rate distortion function based upon results of the encoded  
18   video object plane and upon a memory containing results of one or more  
19   previously encoded video object planes;  
20           after the encoding the MVOs in the video object plane, performs a frame  
21   skipping function; and  
22           transmits, at the estimated available bandwidth, the encoded video object  
23   plane from the sender to the receiver.  
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